AMENDMENTS TO THE SPECIFICATION

Please amend the specification as indicated hereafter. It is believed that the following amendments and additions add no new matter to the present application.

Please replace the paragraph beginning on page 3, line 8 with the following paragraph:

There are also corresponding solving methods related to the specific packet services with time delay sensitivity. For example, US patent application 20030101274 "packet transmission scheduling technique" described a method for scheduling the real time packet service in WCDMA high speed downlink packet access HSDPA system, which judges the weight of packet by the maximum time delay threshold determined by the priority, quantity, time delay and service quality QoS of the packet on the basis of states of the wireless channel. The closer the time delay of data approximates to the maximum time delay threshold, or when the time delay is long enough, the higher transmission priority the data has. However, as for the real time packet service whose data delay ~~excess~~ exceed the maximum time delay threshold, its corresponding data packet will be discarded. The discarding of packet is unavoidable, and with the existing technique, the ~~lost~~ loss ratio of packet can not be further decreased, and there is no related solution provided for the users losing packet. As a result, the ~~lost~~ loss ratio of packet of the users in bad transmission condition remains high, and the communication quality and satisfaction of users may be influenced; Compared with wire transmission, the ~~lost~~ packet loss in wireless transmission often occurs in burst, so special measures are needed to deal with the packet ~~lost~~ loss.

Please replace the paragraph beginning on page 3, line 24 with the following paragraph:

The above methods only consider some of the characters of wireless telecommunication while neglect some others. For example, some of the real time packet services, such as the packet audio and video service, are quite sensitive to time delay jitter, which will greatly affect

the quality of telecommunication. The common solution to this problem is to set data buffer with enough length to compensate the shortcomings of time delay jitter, as in US paten application 20030112796 "Voice and data exchange over a packet based network with fax relay spoofing" and US patent application 20030026275 "Dynamic Jitter buffering for voice-over-IP and other packet-based communication systems". However, in order to control the big time delay jitter, big data buffer has to be set, which will result in the increasing transmission time delay of the date, which when ~~excesses~~ exceeds a certain limitation, the requirement of service quality of wireless telecommunication can not be met. The existing technique does not consider the treatment toward the time delay jitter in packet scheduling under the condition of meeting all the requirements of service quality QoS, therefore the communication quality of some services is destroyed.

Please replace the paragraph beginning on page 4, line 12 with the following paragraph:

The technical problem to be solved in the present invention is to provide a method for scheduling packet in a wireless telecommunication system, for the purposes of decreasing the ~~lost~~ loss ratio of packet and controlling the time delay jitter, therefore optimizing the packet time delay, throughput, time delay jitter and ~~lost~~ loss ratio of packet under the condition of meeting the QoS requirements of all services.

Please replace the paragraph beginning on page 4, line 17 with the following paragraph:

The method for scheduling packet in wireless telecommunication system described in the present invention divides the user packet queues to be transmitted into the user packet queues with ~~lost~~ packet loss and user packet queues without ~~lost~~ packet loss; for the user packet queues with ~~lost~~ packet loss, if a real time ~~lost~~ loss ratio of packet for the user ~~excesses~~ exceeds a predetermined ~~lost~~ loss ratio threshold of packet, terminates a connection to the

user; if the real time ~~lost~~ loss ratio of packet for the user does not ~~excess~~ exceed the predetermined ~~lost~~ loss ratio threshold of packet, schedules the user packet queues according to a volume of the ~~lost~~ loss ratio of packet; for the user packet queues without ~~lost packet~~ loss, schedules according to packet lengths, channel quality states, time delays and time delay jitters.

Please replace the paragraph beginning on page 4, line 26 with the following paragraph:

The present invention, through giving priority to scheduling the users with high ~~lost~~ loss ratio of packet under the condition of existing a certain range of packet ~~lost~~ loss, decreases the ~~lost~~ loss ratio of packet, especially those whose users are in bad transmission condition, thereby ensures the comparative fairness of transmission in unfair transmission conditions. When the ~~lost~~ loss ratio of packet ~~excesses~~ exceeds the threshold, the connections to the users will be terminated temporarily waiting for the improvement of transmission condition. The present invention takes the requirements of packet service for users sensitive to the time delay jitter into full consideration, it controls the time delay jitter to maintain invariable, therefore improves the telecommunication quality of those users.

Please replace the paragraph beginning on page 5, line 20 with the following paragraph:

The core theory of the present invention is that: first, the user packet queues to be transmitted are divided into the user packet queues with ~~lost packet~~ loss and the user packet queues without ~~lost packet~~ loss; for the user packet queues with ~~lost packet~~ loss, if a real time ~~lost~~ loss ratio of packet for the user ~~excesses~~ exceeds a predetermined ~~lost~~ loss ratio threshold of packet, terminates a connection to the user; if the real time ~~lost~~ loss ratio of packet for the user does not ~~excess~~ exceed the predetermined ~~lost~~ loss ratio threshold of packet, schedules the user packet queues according to a volume of the ~~lost~~ loss ratio of packet; for the user

packet queues without ~~lost packet~~ <u>loss</u>, schedules according to packet lengths, channel quality states, time delays and time delay jitters.

Please replace the paragraph beginning on page 6, line 1 with the following paragraph:

As shown in the flowchart of fig.1, first, judge whether the packet queue to be transmitted is vacant or not(step 100), if yes, perform step 108 to exit the scheduling algorithm, if not, obtain the related information required by the scheduling at the beginning of each scheduling period (step 101), the information includes the channel quality states, the length of all the packets to be transmitted, the maximum delay threshold of various services, the delay waiting time of each packet, the real time ~~lost~~ <u>loss</u> ratio of packet of each user, the real time ~~lost~~ <u>loss</u> ratio threshold of packet of each user, the time delay jitter of packet, the time delay jitter threshold of packet.

Please replace the paragraph beginning on page 6, line 21 with the following paragraph:

The real time ~~lost~~ <u>loss</u> ratio of packet of various users is referred as $PL_{i,j}$, wherein i represents one user index, j represents the scheduling period , the value of j is an integer not less than one. The real time ~~lost~~ <u>loss</u> ratio of packet $PL_{i,j}$ can be represented by the proportion of ~~lost packets~~ <u>loss</u> to all of the transmitted packets in a certain period of time, the concrete length of time depends on the statistic period of the whole system, which can usually be selected between 200 ms and 2s, with 200ms is preferable, because the shorter the time is, the faster the response of system is, but since the consumption of system resource will be increased, an overall consideration is necessary. The real time ~~lost~~ <u>loss</u> ratio threshold of packet of each user is referred as PL_{max} , and $PL_{max} > 0$.

Please replace the paragraph beginning on page 7, line 15 with the following paragraph:

After obtaining the above related information, judge whether there are users whose real time ~~lost~~ loss ratio of packet $PL_{i,j}$ is more than 0 (step 102), if yes, judge whether there are users whose real time ~~lost~~ loss ratio of packet $PL_{i,j}$ ~~excesses~~ exceeds the real time ~~lost~~ loss ratio threshold of packet PL_{max} (step 103), for whom, the connection will be terminated, and step 107 will be performed to judge whether the packet queue is vacant; if there is no user whose real time ~~lost~~ loss ratio of packet $PL_{i,j}$ ~~excesses~~ exceeds the real time ~~lost~~ loss ratio threshold of packet PL_{max}, that is, when $PL_{max} \geq PL_{i,j} > 0$, schedule the user packet according to the volume of ~~lost~~ loss ratio of user packet, that is, give priority to scheduling the user packet with big packet ~~lost~~ loss ratio, till all the user packets ~~less~~ than the real time ~~lost~~ loss ratio threshold of packet PL_{max} are scheduled, then perform step 107 to judge whether the packet queue is vacant .

Please replace the paragraph beginning on page 8, line 3 with the following paragraph:

If there is no user whose real time packet ~~lost~~ loss ratio $PL_{i,j}$ is more than 0, perform step 106, that is consider the packet length $l_{i,j}$, channel state $C_{i,j}$, time delay $W_{i,j}$, time delay jitter $Jitter_{i,j}$ etc. comprehensively to schedule the packets in the queue, scheduling with priority can be performed according to the principal of least $\left(W_{max,m} - W_{i,j}\right)\left(Jitter_{max,n} - Jitter_{i,j}\right)l_{i,j}C_{i,j}$, wherein $\left(W_{max,m} - W_{i,j}\right)$ represents the limitation to time delay, $\left(Jitter_{max,n} - Jitter_{i,j}\right)$ represents the limitation to the time delay jitter, $l_{i,j}$ represents the consideration over the packet length , $C_{i,j}$ represents the consideration over the channel state; The principal of least

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$\left(Jitter_{max,n} - Jitter_{i,j}\right)I_{i,j}C_{i,j}/W_{i,j}$ can also be employed for scheduling. After finishing the scheduling, judge whether the packet queue is vacant (step 107).

Please replace the paragraph beginning on page 9, line 1 with the following paragraph:

The scheduling unit 203 buffers the data packet temporarily, calculates the delay waiting time $W_{i,j}$ of each packet, the real time packet ~~lost~~ loss ratio $PL_{i,j}$ of each user, the time delay jitter of packet $Jitter_{i,j}$, and receives the control information S211 transmitted from MAC controller 201, including the QoS requirements of services, the delay threshold $W_{max,m}$, the time delay jitter threshold $Jitter_{max,n}$, the real time packet ~~lost~~ loss ratio threshold PL_{max} of each user, the delay occurred before MAC-hs and the overall power limit of all the HS-DSCH. The uplink signaling S212 from High Speed Downlink Shared Channel (simplified as HS-DSCH) 206 includes such channel state information as the maximum number of transmission bits in the transmission time interval TTI, the modulating manners, the code channel number, etc.. Then the packet date is scheduled in the scheduling unit 203 according to the scheduling method of the present invention.

Please replace the paragraph beginning on page 9, line 16 with the following paragraph:

Fig.3 is the concrete flowchart of packet scheduling by scheduling unit 203 of fig.2. The scheduling method used in HSDPA system divides the levels of priority among the packet with time delay jitter and time delay limitation, the packet only with time delay limitation, and the packet without time delay limitation. Meanwhile, besides scheduling packet according to the period of transmission time interval TTI in terms of time order, the scheduling method also has to schedule the code channel and power within the same TTI period. First, read the packet data to be transmitted through HS-DSCH 206 into the buffer of queue(step 300), since the data may

not be obtained when read, it is necessary to judge whether the buffer queue is vacant (step 301), if is, finish the packet scheduling period of one TTI (step 308), if not, continue to judge whether there is packet service sensitive to time delay (step 302), when there is no such packet service, it indicates that it is packet service without time delay limitation, and perform step 306, when there are packet services sensitive to time delay, then further judge whether there are packet services sensitive to time delay jitter (step 303), if yes, perform step 304, if no, perform step 305. After two judgements of step 302 and step 303, the packet data to be transmitted can be corresponded to packet with time delay jitter and time delay limitation, or packet only with time delay limitation, or packet without time delay limitation, the priority levels of the above three kinds of packet rank from high to low as the packet with time delay jitter and time delay limitation first, the packet only with time delay limitation second, and the packet without time delay limitation third; the higher the priority level of packet service is, the higher the possibility for its data being sent out with fastest speed is. Step 304 is to schedule the packet with time delay jitter and time delay limitation, which can employ the scheduling method of the present invention to assign the code channel and power, with its specific embodiment scheme shown in fig.1. Step 305 is to schedule the packet service without time delay jitter but with time delay limitation, which commonly uses the EDF (Earliest Deadline First) algorithm, that is to give priority to servicing the user packet most approximating to the time threshold. Step 306 is to schedule the packet service without time delay limitation whose priority level is lowest, in which, the ordinary wireless WFQ fair scheduling method can be employed . After performing step (304), step(305), step(306)respectively, perform step(307) to judge whether the code channel assigned in this TTI scheduling period or the overall power used excesses exceeds the upper limit, if yes, finish the packet scheduling period of one TTI(step 308), if not, return to step 300 to continue scheduling the packet service in this TTI scheduling period after re-reading new data.

After finishing step 308, also return to step 300 to schedule the packet service in the next TTI scheduling period after re-reading new data.

For the convenience of the Examiner, a "marked-up" and a "clean" copy of the Application and Abstract is submitted herewith.